

# Data Sampling & Nyquist Theorem

**Richa Sharma**



**Dept. of Physics And Astrophysics  
University of Delhi**

## **Signal :**

*Any physical quantity that varies with time, space, or any other independent variable or variables.*

## **Classification of Signals :**

- Continuous -Time Signals
- Discrete -Time Signals
- Continuous -Valued Signals
- Discrete- Valued Signals

## **Continuous -Time Signals :**

- *defined for every value of time*
- *take on values in continuous interval  $(a, b)$ , where  $a$  can be  $-\infty$  and  $b$  can be  $\infty$ .*
- *can be described by functions of a continuous variables*

## **Discrete -Time Signals :**

- *defined only at certain specific values of time*
- *time instants need not be equidistant, but in practice they are usually taken at equally spaced intervals*

*The values of a continuous-time or discrete-time signal can be continuous or discrete.*

### **Continuous-Valued Signals :**

*If a signal takes on all possible values on a finite or an infinite range it is said to be continuous-valued signals.*

### **Discrete-Valued Signals :**

*If a signal takes on values from a finite set of possible values ,it is said to be a discrete-valued signals.*

## **Signal Processing :**

*It is an area that deals with operations on or analysis of signals, in either discrete or continuous time. Signals of interest can include sounds, images, time-varying measurement values and sensor data.*

**Processing of signals includes the following operations :**

- ✓ Filtering
- ✓ Smoothing
- ✓ Modulation
- ✓ Digitization
- ✓ A variety of other operations

## **Categories of signal processing :**

**Analog Signal Processing:** *Most signals of practical interest ,such as speech,biological signals,seismic signals,radar signals and various communication signals such as video and audio signals ,are analog.*

*Such signals may be processed directly by appropriate analog systems(such as filters) for the purpose of changing their characteristics or extracting the desired information.*

*In such case signal has been processed directly in its analog form.*

**Digital Signal Processing:**

*In this an analog signal is first converted into the digital signal and then processed to extract the desired information.*

# Advantages of Digital over Analog Signal Processing

- Digital system can be simply reprogrammed for other applications/ported to different hardware / duplicated
- Reconfiguring analog system means hardware redesign, testing, verification
- DSP provides better control of accuracy requirements
- Analog system depends on strict components tolerance, response may drift with temperature
- Digital signals can be easily stored without deterioration
- Analog signals are not easily transportable and often can't be processed off-line
- More sophisticated signal processing algorithms can be implemented
- Difficult to perform precise mathematical operations in analog form

## Analog-to Digital Conversion :

➤ Most signal of practical interest, such as

✓ speech

✓ biological signals

✓ seismic signals

✓ radar signals & sonar signals

✓ various communication signals are analog

➤ To process analog signals by digital means ➡ Conversion from analog into digital form



**Analog-to-Digital (A/D) conversion**

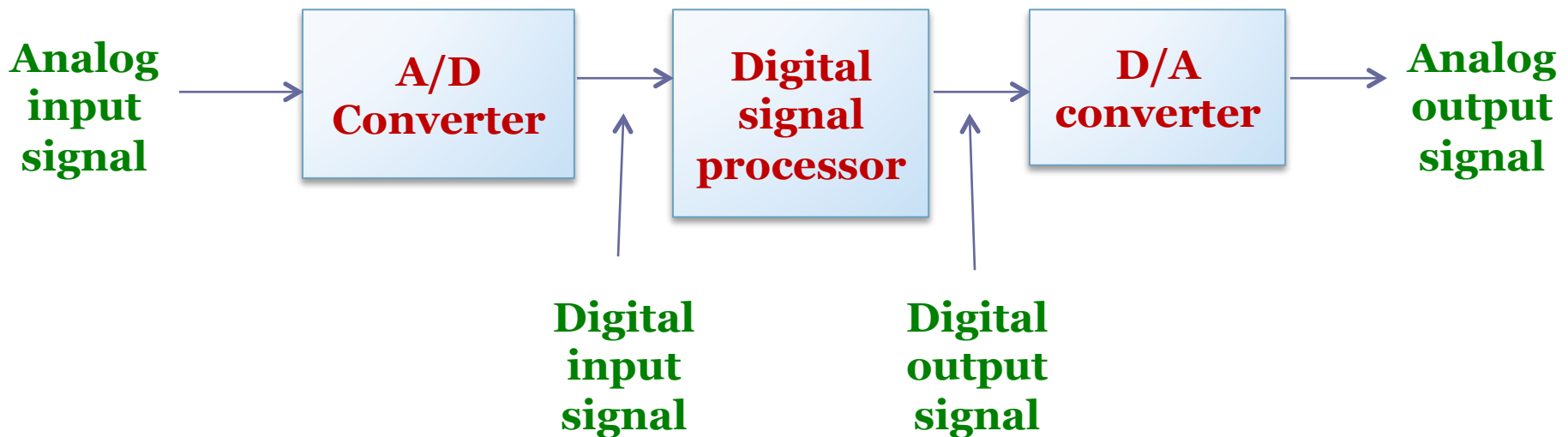


# Digital Signal Processing :

**For a signal to be processed digitally,**

- it must be discrete in time
- Its values must be discrete

**Block diagram of a digital signal processing system**

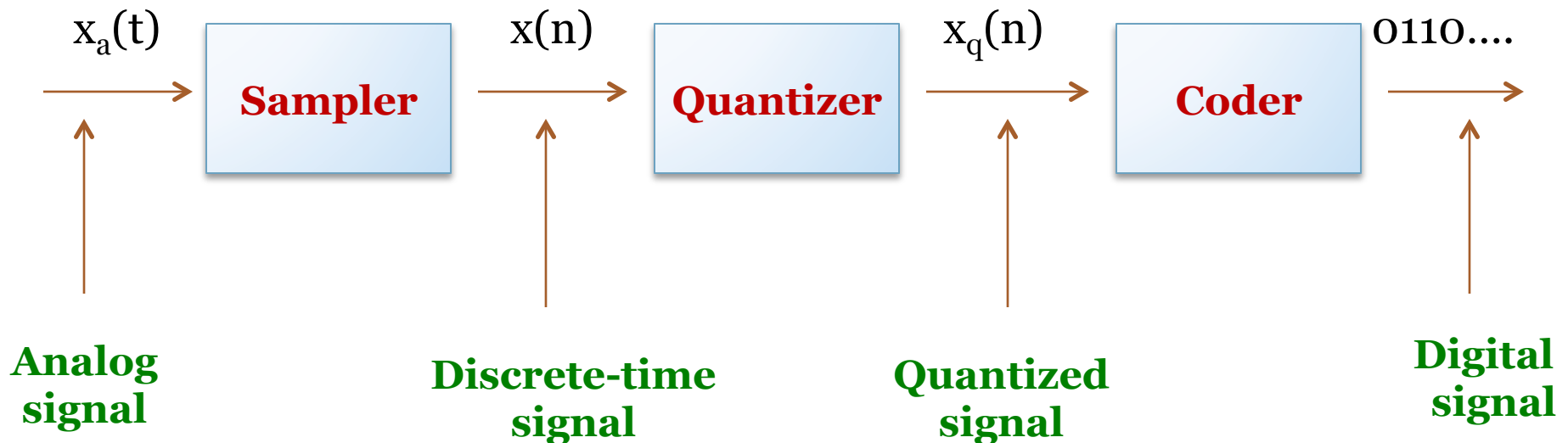


# A/D conversion is a three-step process :

Step 1 : *Sampling*

Step 2: *Quantization*

Step 3: *Coding*



## Step 1 : Sampling of Analog signal

Conversion of continuous-time signal into a discrete-time signal by taking samples of continuous-time signal at discrete time instants.

A continuous time sinusoidal signal is :

$$\mathbf{x_a(t) = A\cos(\Omega t + \theta) , -\infty < t < \infty} \quad (1)$$

Where,

$x_a(t)$  : an analog signal

A : is amplitude of the sinusoid

$\Omega$  : is frequency in radians per seconds(rad/s)

$\theta$  : is the phase in radians

$$\mathbf{\Omega = 2\pi F}$$

A discrete-time sinusoidal signal obtained by taking samples of the analog signal  $x_a(t)$  every  $T$  seconds may be expressed as

$$\mathbf{x(n) = x_a(nT) = A\cos(\omega n + \theta) , \quad -\infty < n < \infty \quad (2)}$$

Where,

$n$  : an integer variable, called sample number

$A$  : is amplitude of the sinusoid

$\omega$  : frequency radians per sample

$\theta$  : is the phase in radians

$$\omega = 2\pi f$$

$f$  : frequency cycles per samples

$T$  is the **sampling interval** or **sampling period**

$F_s = 1/T$  is called the **sampling rate** or the **sampling frequency**  
(Hertz)

➤ **Relationship b/w frequency of analog and digital signal is**

$$\mathbf{f} = \mathbf{F}/F_s$$

➤ **Range of frequency variables**

$$-\infty < F < \infty$$

$$-1/2 < f < 1/2$$

➤ **Frequency of the continuous-time sinusoid when sampled at rate  $F_s$  must fall in the range**

$$- F_s/2 \leq F \leq F_s/2$$

➤ The highest frequency in the discrete signal is  $f = 1/2$  ,

➤ With a sampling rate  $F_s$  , the corresponding highest value of  $F$  is

$$F_{\max} = F_s/2$$



**Sampling introduces an ambiguity**

# Limitations of DSP – Aliasing

Most signals are analog in nature, and have to be sampled



**loss of information**

➤ we only take samples of the signals at intervals and don't know what happens in between



**aliasing**

cannot distinguish between higher and lower frequencies

**Sampling theorem:** to avoid aliasing, sampling rate must be at least twice the maximum frequency component ('bandwidth') of the signal

## Sampling Theorem :

- To avoid ambiguities resulting from aliasing  sampling rate needs to be sufficiently high



$$F_s > 2F_{\max}$$

$F_{\max}$  is the largest frequency component in the analog signal.

- *If the highest frequency contained in the analog signal  $x_a(t)$  is  $F_{\max} = B$  and signal is sampled at a rate*

$$F_s > 2F_{\max}$$

*Then  $x_a(t)$  can be exactly recovered from its sample values*

**The sampling rate  $F_N = 2B = F_{\max}$  is called Nyquist Rate**

## Step 2 : Quantization

*Conversion of a discrete-time continuous valued signal into a discrete-time, discrete-valued (digital) signal by expressing each sample value as a finite number of digits is called quantization.*

- *It is basically an approximation process.*
- *Accomplished by rounding or truncating*

**Quantization Error** :Difference between the quantized value and the actual value

$$e_q(n) = x_q(n) - x(n)$$

Where ,


$x_q(n)$  denote sequence of quantized samples at the output of the quantizer



- **Quantization levels** : *The values allowed in the digital signal are called quantization levels.*
- **Quantization step size** : *The distance between two successive quantization levels is called the Quantization step size or resolution.*  
*Denoted by  $\Delta$ .*
- *The quantizer error  $e_q(n)$  is limited to the range*  
$$-\Delta/2 \leq e_q(n) \leq \Delta/2$$
- *If  $x_{max}$  and  $x_{min}$  represents the maximum and minimum values of  $x(n)$*
- *$L$  is the number of quantization levels*

$$\Delta = (x_{max} - x_{min}) / (L - 1)$$

## Step 3 : Coding of the quantized samples

- *The coding process in A/D converter assign a unique binary number to each quantization level.*
- *In this process, each discrete value  $x_q(n)$  is represented by  $b$ -bit binary sequence.*
- *For  $L$  number of quantization levels we need  $L$  different binary numbers.*
- *With a word length of  $b$  bits   $2^b$  different binary numbers*
- *Hence*

$$2^b \geq L \quad \text{or} \quad b \geq \log_2 L$$

## Applications :

### ➤ **communication systems**

modulation/demodulation, channel equalization, echo cancellation

### ➤ **consumer electronics**

perceptual coding of audio and video on DVDs, speech synthesis, speech recognition

### ➤ **music**

synthetic instruments, audio effects, noise reduction

### ➤ **medical diagnostics**

magnetic-resonance and ultrasonic imaging, computer tomography, ECG, EEG, MEG, AED, audiology

### ➤ **geophysics**

seismology, oil exploration

➤ **astronomy**

VLBI, speckle interferometry

➤ **experimental physics**

sensor-data evaluation

➤ **aviation**

radar, radio navigation

➤ **security**

steganography, digital watermarking, biometric identification, surveillance systems, signals intelligence, electronic warfare

➤ **engineering**

control systems, feature extraction for pattern recognition



Thank You